

REMARKS

As set forth in the attached LETTER TO THE OFFICIAL DRAFTSPERSON, Fig. 3 of the drawings has been amended to correct the text in the decision box referenced as S10 to replace the text "PITCH FLAG = 0?" to --PITCH LAG = 0?-- as shown on the attached red-lined drawing. The proposed correction to the drawing is shown in red ink in the attached sketch of Fig. 3. "PITCH LAG" is discussed at page 19, lines 22-23.

Claims 1-22 are pending. Claims 1-6, 8, 10-14 and 17 have been amended to place them in better form to provide proper antecedent basis for the elements recited. Claim 1 has been amended to recite "a speech signal encoder for compressing input speech signals by digital signal processing for high quality voice transmission at a low bit rate and for producing analytic parameters from the input speech signal" which is supported by claim 1 of the application, as originally filed, and at p. 7, ll. 11-18 of the specification as originally filed. Claim 1 further recites a "noise domain detection means supplied with patterns of analytic parameters produced by said speech signal encoder during compression of the input speech signals for determining a noise domain in which only noise exists in the input speech signal" which finds support in the specification, as originally filed, at p. 4, ll. 1-3 and ll. 16-20. Claim 1 has also been amended to recite a "noise level detecting means for detecting a noise level of the input speech signal in the noise domain" which finds support in the specification, as originally filed, at p. 4, ll. 3-25. Finally, claim 1 now recites a "means for controlling a volume of the corresponding received sound responsive to the

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noise level detected by said noise domain detection means" which is supported in the specification, as originally filed, at p. 11, ll. 14 to p. 12, l. 21.

Claims 4 and 11 have been amended to recite a noise domain detection means which "employs a pitch lag as one of the analytic parameters for each frame and deems a frame to be a noise domain if the pitch lag is zero." The claims as amended find support in the specification, as originally filed, at p. 19 , ll. 22 to p. 20, ll. 4.

Claims 1-22 are pending. Previously, claims 1 and 17-22 were rejected under 35 U.S.C. §103 over Borth et al. in view of Hung et al. Claims 2-16 were rejected under 35 U.S.C. §103 over Borth et al. in view of Hung et al. and further in view of Gerson et al. These rejections are respectfully traversed.

The rejections are similar to, and incorporate the same contentions made in the first office action in parent case Application No. 08/263,125. This preliminary argument, therefore, incorporates the same rebuttal arguments made in response to the first office action and which have not been addressed by the Examiner in the final rejection nor in the advisory action. To the extent that the Examiner maintains these grounds of rejection, clarification would be appreciated.

In particular, the Examiner does not address the salient feature, present in all of the claims, that the present device controls the volume of an audibly reproduced signal **transmitted from** a remote transmitter so as to overcome **background noise surrounding the user** of the portable telephone. The Examiner may be assuming that what is attenuated at the portable telephone is either the ambient noise or

the sound input into a microphone of the portable telephone. Neither is the case.

The rejection of claims 1 and 17-22 under §103 over Borth et al. in view of Hung et al.

The present invention finds its application in, e.g. a portable telephone where it is desired to control the volume of an audibly reproduced signal **transmitted** from a remote transmitter so as to overcome background noise surrounding the user of the portable telephone. Thus, it is an object of the present invention to control the volume of the reproduced speech signal **sent** from a remote transmitter to the portable telephone to be loud enough to be heard over the ambient noise surrounding the user of the portable telephone. See the specification at pages 2-3, page 11, lines 21-23, and page 12, lines 17-21. This feature is clearly set forth in the claims.

Borth et al. does not teach or suggest both a transmitting and a receiving apparatus nor controlling the volume of **reproduced sound signals transmitted to a receiver as a function of detected ambient noise surrounding the receiver**. At best, Borth et al. only suggests a noise suppression circuit for "controlling the volume" of the speech entering the microphone of a portable telephone, i.e. by subtracting out noise, as a function of noise detected in the signal received by the microphone of the portable telephone. But Borth et al. does not control the volume of reproduced sound signals being **received** from a remote transmitter. In particular, Borth et al. does not teach or suggest **any** of a **speech signal encoder** for compressing input speech signals by digital signal processing for high quality

voice transmission at a low bit rate, **a transmitting and receiving circuit** for transmitting the compressed speech signals output by the speech signal encoder and for receiving compressed speech signals transmitted from another transmitter and reproducing a corresponding received sound, **noise domain detection means** supplied with patterns of analytic parameters produced by said speech signal encoder during compression of the input speech signals for determining a noise domain in which only noise exists in the input speech signal, **noise level detecting means** for detecting a noise level of the input speech signal in the noise domain, or **means for controlling a volume of the corresponding received sound** responsive to the noise level detected by the noise domain detection means, all of which are required by claim 1 (and dependent claims 2-16, inclusive). Hung et al.'s VSELP encoder, even if there is some suggestion to make the combination promoted by the Examiner, would still not combine to make a device covered by the claims.

Further, Borth et al. do not teach or suggest a transmitter and a receiver comprising noise level detection means for detecting a sound signal level entering a transmitting microphone as a noise level when there is no transmitting speech input at the transmitter, and control means for **controlling a volume of a sound reproduced from a compressed speech signal from another transmitter responsive to the noise level** detected by the noise level detection means, as required by claims 17-22, inclusive. Again, Hung et al.'s VSELP encoder, even if combined with Borth et al., would not produce a combination which meets the claims.

The Examiner is required to consider the invention as a whole when applying §103. In applying §103, the decision-maker must ascertain the differences between the prior art and the claims at issue. In failing to take account all of the recited elements of Applicants' claims above, the Examiner has made an improper §103 analysis of the obviousness of the above claims. Reconsideration of Applicants' claims in their entirety will demonstrate that it was not obvious from the art relied upon by the Examiner to attempt the solution invented by the Applicants.

The rejection of claims 2-16 under 35 U.S.C. §103 over Borth et al. in view of Hung et al. and further in view of Gerson et al.

The Examiner has again rejected claims 2-16 under 35 U.S.C. §103 as being unpatentable over Borth et al. in view of "Vector Sum Excited Linear Prediction (VSELP) Speech Coding At 8 KBPS", Ira A. Gerson and Mark A. Jasiuk ("Gerson et al."). It is respectfully submitted that Applicants' invention as set forth in claims 2-16 is not obvious in view of the cited prior art.

Claims 2-9

Claim 2 requires that the noise domain detection means employs a first-order linear prediction encoding coefficient as one of the analytic parameters for each frame of a plurality of frames and deems a frame to be the noise domain if the first-order linear prediction encoding coefficient is smaller than a pre-set threshold. The Examiner's statement in the first Office Action at page 7, lines 19-24, that "It would have been obvious to a person of ordinary skill in the art at the time the invention was made to use linear prediction encoding

coefficients as the analytic parameter to determine whether a noise domain is present because this achieves high speech quality while maintaining reasonable complexity (Pg 1, left col, 4th paragraph)" is a non-sequitur and does not create a prima facie case of obviousness. It is a purely conclusory statement, unsupported by the reference.

Gerson et al. teach speech coding using VSELP parameters, however, neither the Applicants nor the undersigned could find any disclosure in Gerson et al. which teaches or suggests *using a speech signal encoder or any analytic parameters produced by it to detect noise or to control the volume of a received sound signal*. The Applicants respectfully disagree that the portions of Gerson et al. cited by the Examiner support such a contention. There appears to be nothing in those portions of the Gerson et al. text which disclose or suggest detecting noise at all. The Examiner is respectfully requested to clarify which portions of Gerson et al. the Examiner is relying on to support his contention that Gerson et al. teaches or suggests noise domain detection means. All that Gerson et al. appear to disclose is a VSELP encoder and nothing more. The leap to the conclusion that it would be obvious to employ such a VSELP encoder to detect the noise domain is purely in the hindsight of the present specification.

Thus, the rejection of claim 2 (and dependent claims 3-9) must fail on two counts. First, as argued above, Borth et al. do not teach or suggest controlling the volume of a reproduced received sound signal as a function of the ambient noise (as more particularly detailed in the claim language paraphrased above in regard to claim 1) nor do Gerson et al. teach or suggest using a first-order linear

prediction encoding coefficient as an analytic parameter for each frame of a plurality of frames and deeming a frame to be the noise domain if the first-order linear prediction encoding coefficient is smaller than a pre-set threshold, as is required by claims 2-9.

Claims 3 and 10

Claims 3 and 10 require that the noise domain detection means employs a pitch gain indicating the intensity of pitch components as one of the analytic parameters for each frame and deems a frame to be the noise domain if the pitch gain is within a pre-set range.

The Examiner cites Gerson et al.'s statements that his VSELP speech decoder uses, as one of three excitation sources, "a long term ('pitch') predictor state" and a "pitch" prefilter as meeting this limitation. The Applicants disagree. There is nothing in Gerson et al. which teaches or suggests determining a noise frame by using pitch gain as an indication of anything, much less as indicating the intensity of pitch components as an analytic parameter for each frame to determine that a frame is the noise domain if the pitch gain is within a pre-set range, as is required by the claims 3 and 10. At best, all that Gerson et al. disclose regarding pitch is that a long term filter state (as a pitch predictor) can be used as one of three excitation sources and that the combined excitation signal is processed by an adaptive "pitch" prefilter to enhance periodicity of the excitation signal. See page 4, paragraph VII.

Claims 4 and 11

Claims 4 and 11 require that the noise domain detection means employs a pitch lag as one of the

analytic parameters for each frame and deems a frame to be the noise domain if the pitch lag is zero.

There is nothing in Gerson et al. which teaches or suggests determining a noise frame by using a lag in the pitch frequency as an indication of anything, much less as indicating the lag of pitch frequency as an analytic parameter for each frame to determine that a frame is the noise domain if the pitch lag is zero, as is required by the claims 4 and 11.

Claims 5 and 12

Claims 5 and 12 require that the noise domain detection means employs a frame power as one of the analytic parameters for each frame and deems a particular frame to be the noise domain if the frame power for the particular frame is smaller than a pre-set threshold. Again, there is nothing in Gerson et al. which teach or suggest such a feature. Gerson et al. quantizes three excitation gains in two stages. In the first stage, the average speech energy is coded once per frame. This, ultimately, allows the average energy to be factored out so that the three gains can be vector quantized efficiently. See page 3, right hand column, first full paragraph. However, there is nothing in this process which suggests a modification of Borth, et al. in view of Hung, et al. to detect a noise domain by employing a frame power as the analytic parameter for each frame and deeming a particular frame to be the noise domain if the frame power for the particular frame is smaller than a pre-set threshold.

Claims 6, 7 and 13, 14

Claims 6, 7 and 13, 14 require that if an amount of change of the frame power between a current frame and a past frame exceeds a pre-set threshold, the noise domain detection means deems the current frame

to be a speech domain, even if the current domain is the noise domain. The Examiner finds this suggested by Gerson et al. at page 4, section IV wherein Gerson et al. describes solving 560 simultaneous equations which result from taking partial derivatives of the total normalized weighted error function with respect to each sample of each basis vector and setting them equal to zero. Because the code frames are not independent, the procedure is iterated in a closed loop fashion. The Examiner apparently contends that solving 560 simultaneous equations in an iterative, closed loop fashion "implies that subtraction is being done between current and past iteration of the frame energies." It is respectfully submitted that this is such a non-sequitur that it is difficult to formulate a response. Iterative solutions of simultaneous equations does not imply determining if an amount of change of the frame power between a current frame and a past frame exceeds a pre-set threshold, and if so, deeming the current frame to be a speech domain, even if the current domain is the noise domain.

Claims 8 and 15, and dependent claims 9 and 16

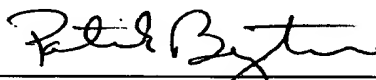
Claims 8 and 15, and consequently dependent claims 9 and 16, respectively, require a noise level detection means which performs filtering on a noise level output of the noise domain detected by the noise domain detection means. The Examiner points to Gerson et al.'s pitch prefilter in Figure 1 as satisfying this limitation. However, the elements of Gerson et al. prior to the pitch prefilter, i.e. the elements to the left of the pitch prefilter in Figure 1, do not constitute a noise domain detection means. They are separate signal sources, not a signal detector. Thus, the pitch prefilter can not serve a

function of filtering a noise level of a **detected noise domain** output from a noise domain **detection** means.

For the foregoing reasons it is submitted that all of the claims are now in condition for allowance and the Examiner's early examination and consideration are respectfully requested.

Respectfully submitted,
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